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**IN THE CLAIMS:**

Please amend the claims as follows:

1. (Currently Amended) A method, comprising the steps of:

receiving, a first voice traffic at a first Voice over Internet Protocol (VoIP) gateway;

determining whether a destination is serviced by a second VoIP gateway;

multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and

transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway, wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport ~~Packet~~ Protocol (RTP) packet,

wherein said modified RTP packet comprises at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

2. (Previously presented) The method of claim 1, wherein said voice traffic is received within the payload portions of User Datagram Protocol (UDP)/Internet Protocol (IP) packets.

3-5. (Cancelled)

6. (Previously presented) The method of claim 1, wherein said Header Error Check field performs one bit error correction.

7. (Previously presented) The method of claim 1, further comprising the step of communicating messages between said VoIP gateway and said second VoIP gateway.

8. (Original) The method of claim 7, wherein said first VoIP gateway communicates an Open Logical Channel message to said second VoIP gateway including said VoIP gateway's port number and Call Identifier of the calling party.

9. (Original) The method of claim 8, wherein in response to said Open Logical Channel message said second VoIP gateway communicates an Open Logical Channel message including said second VoIP gateway's port number and Call Identifier for the called party.

10. (Original) The method of claim 7, wherein in response to a caller terminating a call said VoIP gateway communicates a Close Logical Channel message including said VoIP gateway's port number and said Call Identifier of the calling party to said second VoIP gateway.

11. (Original) The method of claim 10, wherein in response to said Close Logical Channel message said second VoIP gateway communicates a Close Logical Channel ACK message including said second VoIP gateway's port number and said Call Identifier of the called party.

12. (Original) The method of claim 1, wherein said step of determining is made utilizing a gatekeeper.

13. (Currently Amended) In a communication system for transporting voice traffic over an Internet Protocol (IP) network to a destination, apparatus comprising:

a first Voice over Internet Protocol (VoIP) gateway, for receiving a first voice traffic;

said first VoIP gateway determining whether said destination is serviced by a second VoIP gateway;

said first VoIP gateway multiplexing said first voice traffic with a second voice traffic, if said second voice traffic is being provided to said second VoIP gateway;

said first VoIP gateway transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway, wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport ~~Packet~~ Protocol (RTP) packet,

wherein said modified RTP packet comprises at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

14. (Original) The apparatus of claim 13, wherein said voice traffic is received within the payload portions ofUser Datagram Protocol (UDP)/Internet Protocol (IP) packets.

15-17. (Cancelled)

18. (Previously presented) The apparatus of claim 13, wherein said Header Error Check field performs one bit error correction.

19. (Original) The apparatus of claim 18, further comprising the step of communicating messages between said VoIP gateway and said second VoIP gateway.

20. (Original) The apparatus of claim 19, wherein said first VoIP gateway communicates an Open Logical Channel message to said second VoIP gateway including said VoIP gateway's port number and Call Identifier of the calling party.

21. (Original) The apparatus of claim 20, wherein in response to said Open Logical Channel message said second VoIP gateway communicates an Open Logical Channel message including said second VoIP gateway's port number and Call Identifier for the called party.

22. (Original) The apparatus of claim 21, wherein in response to a caller terminating a call said VoIP gateway communicates a Close Logical Channel message including said VoIP gateway's port number and said Call Identifier of the calling party to said second VoIP gateway.

23. (Original) The apparatus of claim 22, wherein in response to said Close Logical Channel message said second VoIP gateway communicates a Close Logical Channel ACK message including said second VoIP gateway's port number and said Call Identifier of the called party.

24. (Original) The apparatus of claim 13, wherein a gatekeeper is used to determine whether said second VoIP gatekeeper services said destination.

25. (Currently Amended) A first Voice over Internet Protocol (VoIP) gateway for transporting voice traffic over an Internet Protocol (IP) network to a destination, comprising:

a processor; and

a storage device coupled to said processor and including instructions for controlling said processor, said processor operative with said instructions to:

receive a first voice traffic at said first VoIP gateway;

determine whether said destination is serviced by a second VoIP gateway;

multiplex, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and

transport said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway, wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport ~~Packet~~ Protocol (RTP) packet,

wherein said modified RTP packet comprises at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

26. (Original) A Voice over Internet Protocol (VoIP) gateway for transporting voice traffic over an Internet Protocol (IP) network to a destination as in claim 25, wherein a gatekeeper is used to determine whether said destination is serviced by said second VoIP gateway.

27. (Currently Amended) A first Voice over Internet Protocol (VoIP) gateway, for transporting voice over an Internet Protocol (IP) network, to a destination, comprising:

means for receiving a first voice traffic at said first VoIP gateway;

means for determining whether said destination is serviced by a second VoIP gateway;

means for multiplexing, at said first VoIP gateway, said first voice traffic with a second voice traffic if said second voice traffic is being provided to said second VoIP gateway; and

means for transporting said multiplexed voice traffic to said second VoIP gateway utilizing a plurality of transport packets, responsive to an affirmative determination that said destination is serviced by said second VoIP gateway, wherein said transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport ~~Packet~~ Protocol (RTP) packet,

wherein said modified RTP packets comprise at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

28. (Original) The VoIP gateway of claim 27, wherein said voice traffic is received within the payload portions ofUser Datagram Protocol (UDP)/Internet Protocol (IP) packets.

29-31. (cancelled)

32. (Previously presented) The VoIP gateway of claim 27, wherein said Header Error Check field performs one bit error correction.

33. (Currently Amended) A method, comprising the steps of:

receiving a voice traffic at a first Voice over Internet Protocol (VoIP) gateway;

transporting the voice traffic to a second VoIP gateway utilizing a plurality of transport packets if a destination of the voice traffic is serviced by the second VoIP gateway, wherein the transport packets are User Datagram Protocol (UDP)/Internet Protocol (IP) packets, and wherein said UDP/IP packets transport at least one modified Real-time Transport ~~Packet~~ Protocol (RTP) packet,

wherein said modified RTP packets comprise at least one of:

a Payload field for containing a voice traffic;

a Call Identifier field for identifying a caller;

a Length Indicator field for identifying the size of the payload field; and

a Header Error Check field for identifying errors in the Call Identifier field and the Length Indicator field.

34. (Cancelled)